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(54) **ADJUSTING A DATA RATE OF A DIGITAL AUDIO STREAM BASED ON DYNAMICALLY DETERMINED AUDIO PLAYBACK SYSTEM CAPABILITIES**

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G10L 19/002 (2013.01)

(52) **U.S. Cl.**
CPC ***G10L 19/24*** (2013.01); ***G10L 19/002*** (2013.01)

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CPC G10L 19/24; G10L 19/002; G10L 25/69; H04R 29/00
USPC 700/94; 381/56–60
See application file for complete search history.

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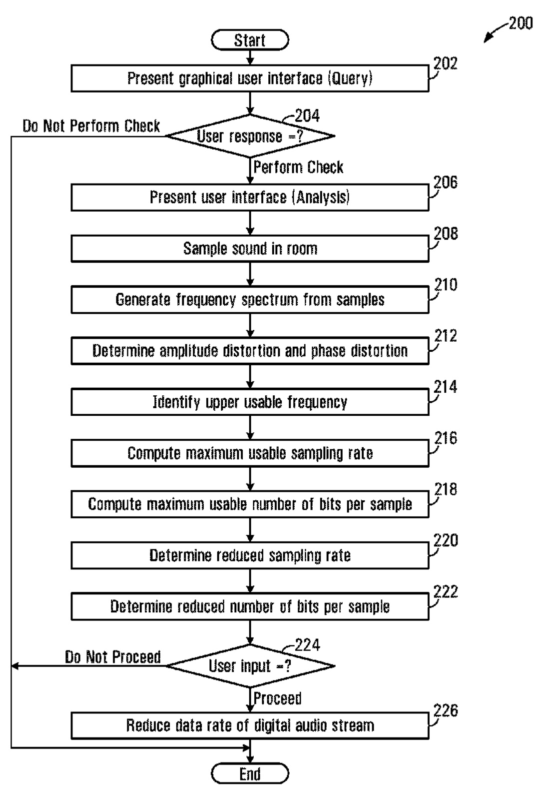
(57) **ABSTRACT**

A computing device may be configured to output a digital audio stream to an audio playback system for rendering as sound over speakers. The sound may be sampled. Based at least in part on a quality of the sampled sound, the data rate of the digital audio stream may be reduced by reducing a sampling rate and/or by reducing a number of bits per sample. A reduced sampling rate may be determined based on a computed maximum sampling rate of the audio playback system, and/or a reduced number of bits per sample may be determined based on a computed maximum number of bits per sample of the audio playback system. The maximum usable sampling rate and maximum usable number of bits per sample may be determined based on an upper usable frequency within a frequency spectrum of the sampled sound.

20 Claims, 3 Drawing Sheets



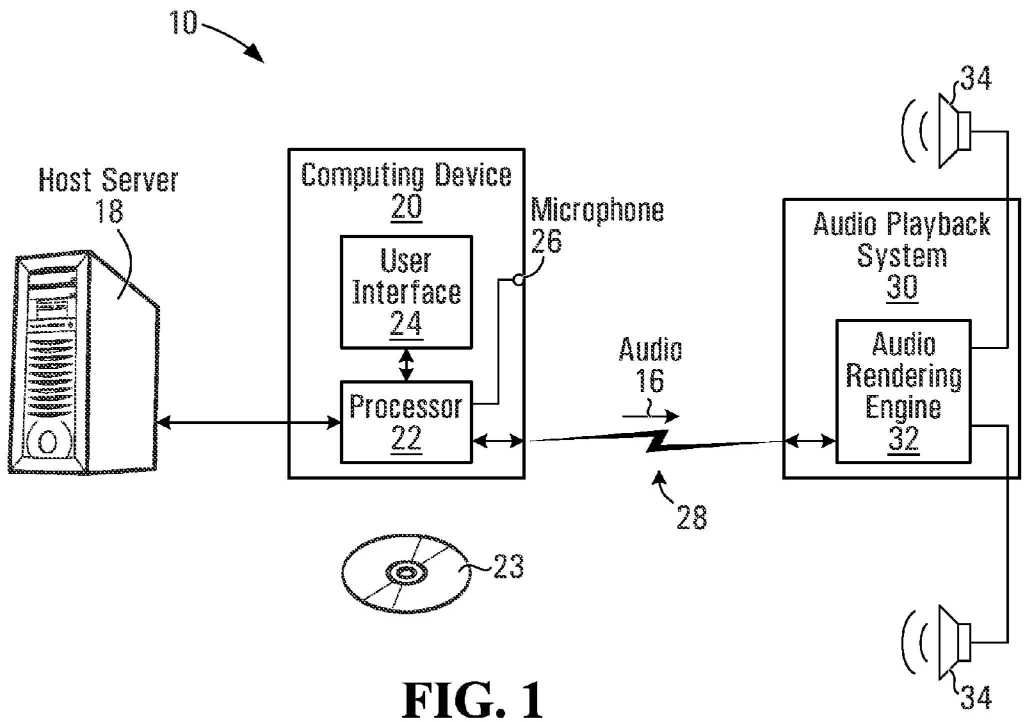

10
Host Server
18
Computing Device
20
User
Interface
24
Processor
22
Microphone
26
Audio
16
28
23
Audio Playback
System
30
Audio
Rendering
Engine
32
34
34

FIG. 1

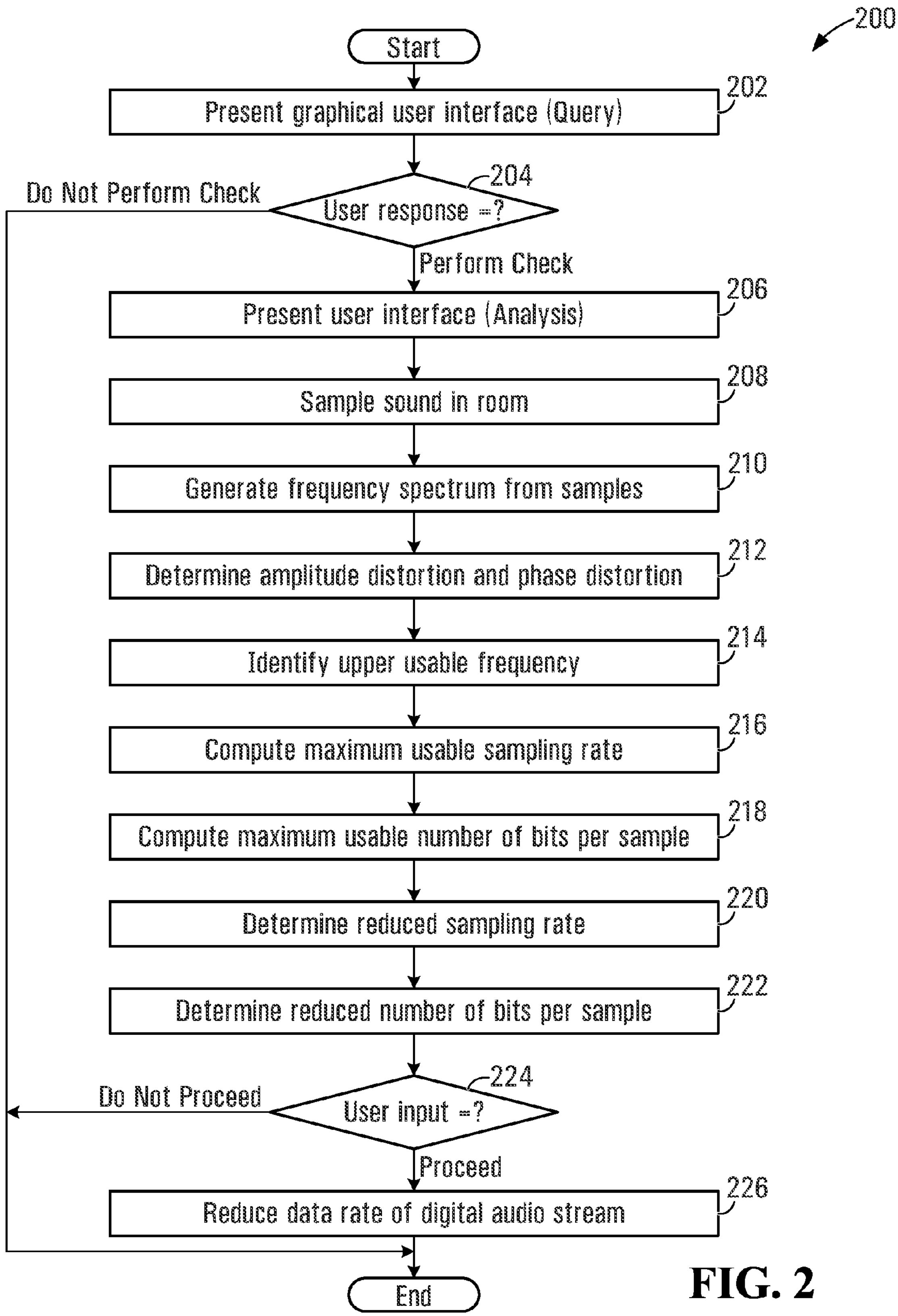

200
Start
Present graphical user interface (Query)
202
User response =?
204
Do Not Perform Check
Perform Check
Present user interface (Analysis)
206
Sample sound in room
208
Generate frequency spectrum from samples
210
Determine amplitude distortion and phase distortion
212
Identify upper usable frequency
214
Compute maximum usable sampling rate
216
Compute maximum usable number of bits per sample
218
Determine reduced sampling rate
220
Determine reduced number of bits per sample
222
User input =?
224
Do Not Proceed
Proceed
Reduce data rate of digital audio stream
226
End

FIG. 2

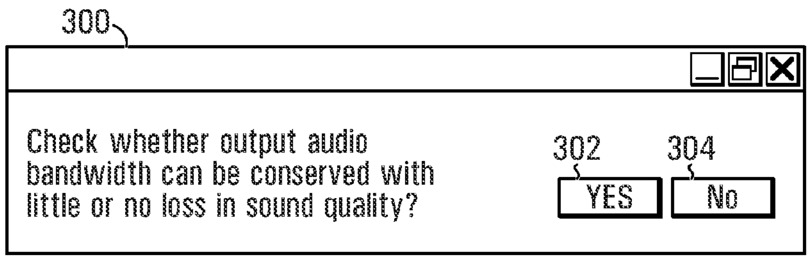

300
Check whether output audio bandwidth can be conserved with little or no loss in sound quality?
302
304
YES
No

FIG. 3

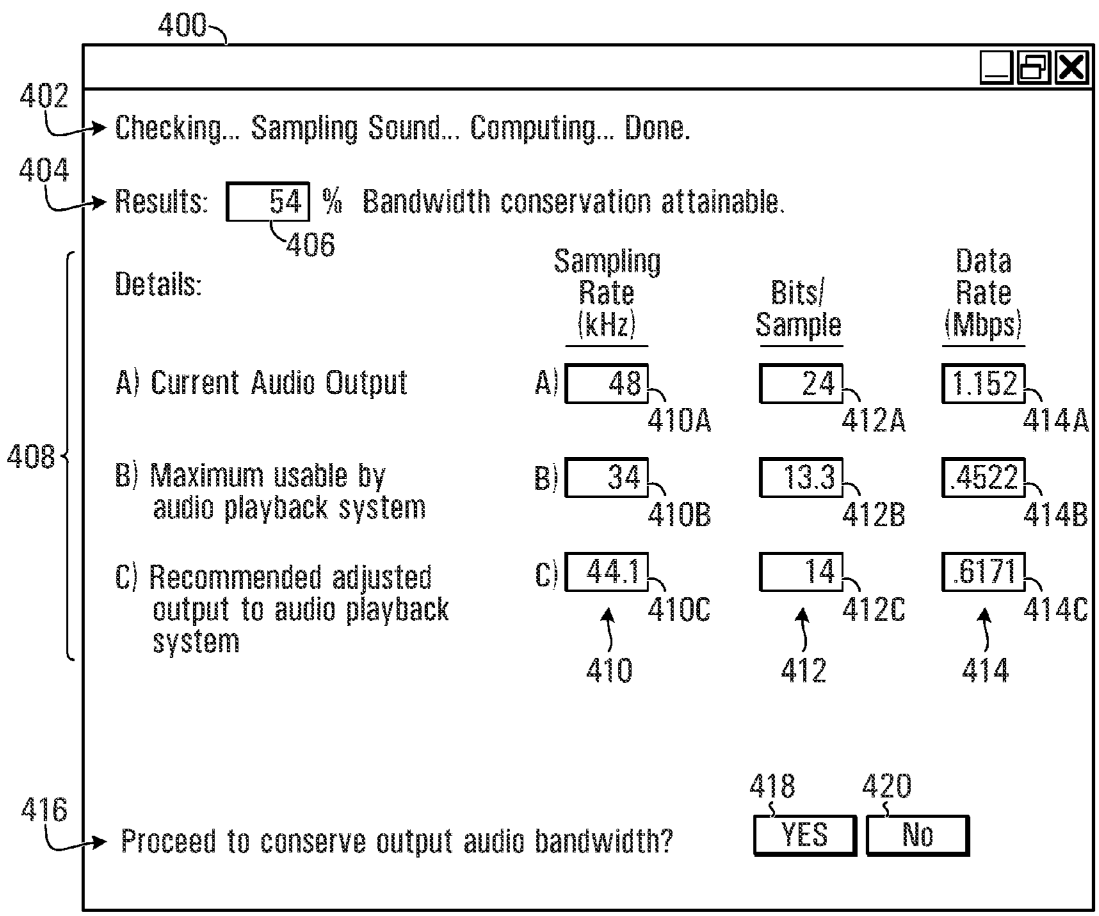

400
402
Checking... Sampling Sound... Computing... Done.
404
Results: 54 % Bandwidth conservation attainable.
406
408
Details:
Sampling Rate (kHz)
Bits/ Sample
Data Rate (Mbps)
A) Current Audio Output
A) 48 410A
24 412A
1.152 414A
B) Maximum usable by audio playback system
B) 34 410B
13.3 412B
.4522 414B
C) Recommended adjusted output to audio playback system
C) 44.1 410C
14 412C
.6171 414C
410
412
414
416
Proceed to conserve output audio bandwidth?
418
420
YES
No

FIG. 4

ADJUSTING A DATA RATE OF A DIGITAL AUDIO STREAM BASED ON DYNAMICALLY DETERMINED AUDIO PLAYBACK SYSTEM CAPABILITIES

FIELD OF TECHNOLOGY

The present disclosure pertains to audio playback systems and associated devices, and more specifically to adjusting a data rate of a digital audio stream based on dynamically determined audio playback system capabilities.

BACKGROUND

Audio playback systems, such as stereo receivers, Audio/Video (AV) receivers, portable stereos, amplified speaker systems, and the like, may receive an audio stream from any one of a number of audio sources and render the audio stream as sound over speakers. The audio stream may be an uncompressed digital audio stream, such as a Linear Pulse-Code Modulation (LPCM) encoded stream, or a compressed digital audio stream that has been created using either a lossless compression technology, such as the Free Lossless Audio Codec (FLAC), or a lossy compression technology, such as MPEG-1 or MPEG-2 Audio Layer III (MP3).

Rendering of audio at the audio playback system may entail processing the audio stream in various ways, e.g., to improve, enhance, or customize the sound that is generated by the audio playback system. This rendering may entail the use of a Digital Signal Processor (DSP), which may be an Application Specific Integrated Circuit (ASIC, i.e. a chip) that is hardwired within the audio playback system. For example, a DSP chip may provide alternative audio field simulations for generating different audio effects such as "hall," "arena," "opera" and the like, which simulate, e.g. using surround sound and echo effects, audio playback in different types of venues.

The nature of the audio rendering that is performed by the audio playback system may be predetermined and fixed, or may be user-selectable from only a finite number of predetermined alternatives. This may be due to limited or fixed audio processing capabilities of the ASIC DSP, or other components, that may be used at the audio playback system for rendering audio. Sound quality may vary depending upon the audio rendering that is performed, the physical attributes of the speakers over which the sound is played (e.g. size, number, configuration, wattage, etc.), and/or the physical characteristics of a room in which the sound is played (e.g. anechoic quality or amount of reverberation).

SUMMARY

In one aspect, there is provided a method of adjusting a data rate of a digital audio stream, the method comprising: sampling sound generated, from a digital audio stream, by an audio playback system; based at least in part on a quality of the sampled sound, reducing the data rate of the digital audio stream by performing either one or both of: reducing a sampling rate of the digital audio stream to a reduced sampling rate; and reducing a number of bits per sample of the digital audio stream to a reduced number of bits per sample.

In another aspect, there is provided a computing device configured for outputting a digital audio stream to an audio playback system for rendering as sound over speakers, the computing device comprising a processor, the processor operable to adjust a data rate of the digital audio stream by: sampling the sound generated, from the digital audio stream, by the audio playback system; and based at least in part on a quality of the sampled sound, reducing the data rate of the digital audio stream by performing either one or both of: reducing a sampling rate of the digital audio stream to a reduced sampling rate; and reducing a number of bits per sample of the digital audio stream to a reduced number of bits per sample.

In another aspect, there is provided a tangible machine-readable medium storing instructions that, upon execution by a processor of a computing device, the computing device configured to output a digital audio stream to an audio playback system for rendering as sound over speakers, cause the processor to: sample the sound generated, from the digital audio stream, by the audio playback system; and based at least in part on a quality of the sampled sound, reduce the data rate of the digital audio stream by performing either one or both of: reducing a sampling rate of the digital audio stream to a reduced sampling rate; and reducing a number of bits per sample of the digital audio stream to a reduced number of bits per sample.

BRIEF DESCRIPTION OF DRAWINGS

In the figures which illustrate example embodiments:

FIG. **1** is a block diagram illustrating an example system;

FIG. **2** contain a flow chart illustrating operation of a computing device in the system of FIG. **1**; and

FIGS. **3** and **4** illustrate graphical user interfaces that may be presented by the computing device whose operation is illustrated in FIG. **2**.

DETAILED DESCRIPTION

FIG. **1** illustrates an example system **10** comprising a computing device **20** and an audio playback system **30** interconnected by a communications link **28**. The computing device **20** outputs a digital audio stream over communications link **28** to the audio playback system **30**, which renders the digital audio stream as sound over speakers. The present disclosure describes an exemplary approach for adjusting, based on dynamically determined capabilities of the audio playback system **30**, the data rate of the digital audio stream that, in at least some embodiments, may allow bandwidth to be conserved with little or no sound quality being lost.

Computing device **20** is an electronic computing device that is capable of outputting a digital audio stream over a communications link **28**. The audio stream may be uncompressed or compressed and may be in one of a wide variety of formats. Examples of uncompressed audio formats include LPCM, Waveform Audio File (WAV), Audio Interchange File Format (AIFF), or AU. Examples of compressed audio formats generated using a lossless compression technology include FLAC, WavPack (.WV extension), True Audio (TTA), Adaptive Transform Acoustic Coding (ATRAC) Advanced Lossless, Apple® Lossless (.M4A extension), MPEG-4 Scalable to Lossless (SLS), MPEG-4 Audio Lossless Coding (ALS), MPEG-4 Direct Stream Transfer (DST), Windows Media Audio (WMA) Lossless), and Shorten (SHN). Examples of compressed audio formats generated using a lossy compression technology include MP3, Dolby™ Digital, Advanced Audio Coding (AAC), ATRAC and WMA Lossy. Other formats not expressly enumerated herein, or as yet unreleased, are also contemplated.

The computing device **20** may be one of a wide variety of different types of electronic devices, such as a desktop computer, PC, laptop, handheld computer, tablet, netbook, mobile device, smartphone, portable music player, video game console, or other type of computing device. As such, computing device **20** may either be a general purpose device (e.g. a general purpose computer) or a special purpose device (e.g. a music player). The computing device **20** comprises at least one processor **22** in communication with volatile and/or non-volatile memory and other components, most of which have been omitted from FIG. **1** for the sake of brevity. The processor **22** may be one of a number of different types of processors, such as a Central Processing Unit (CPU), Accelerated Processing Unit (APU), Graphics Processing Unit (GPU) or other type of processor. The processor **22** may be singular or multiple (e.g. a plurality of parallel processors working in unison). Omitted components of computing device **20** may include a network interface, which may be used if the communications link **28** comprises a network (e.g. a wireless or wired network).

The computing device **20** of FIG. **1** includes a user interface **24**. The user interface **24** comprises a display and a user input mechanism, neither of which is expressly illustrated in FIG. **1**. The display could be virtually any type of display, such as a Liquid Crystal Display (LCD), Light Emitting Diode (LED) display, Organic LED (OLED) display, Plasma display, Cathode Ray Tube (CRT), or others. The user input mechanism may also be of virtually any type, including but not limited to a keyboard, pointing device (e.g. mouse, trackball, trackpad or stylus), touchscreen, or voice-activated input mechanism. As will become apparent, the user interface **24** may optionally be used to control operation of the computing device **20** for adjusting the data rate of the digital audio stream output over communications link **28**.

The example computing device **20** of FIG. **1** also has a microphone **26**. The microphone **26** may be permanently mounted in the housing of the computing device **20** or may be removably interconnected to the computing device **20**, e.g. by being plugged into an audio input jack (such as an RCA mini or 3.5 mm jack for example). The microphone **26** is used for the exemplary data rate adjustment described herein.

The operation of computing device **20** as described herein may be wholly or partly governed by software or firmware loaded from a non-transitory, tangible machine-readable medium **23**, such as an optical storage device or magnetic storage medium for example. The medium **23** may store instructions executable by the processor **22** or otherwise governing the operation of computing device **20**.

The digital audio stream that is output by the computing device **20** may be based on an audio stream received from an upstream host server **18**. The term "upstream" is in relation to the general flow of an audio stream throughout the system **10**, which is from computing device **20** to audio playback system **30**. The host server **18** may for example may be a commercial server operated by an online digital media store (e.g. the iTunes™ Store), an internet service provider or other entity. Alternatively, the host server **18** may be another type of internet-based or wide area network based server, enterprise server, home network-based server or otherwise. These examples are for illustration only and are non-limiting. In some embodiments, the digital audio stream that is output by the computing device **20** may originate at the device **20**, with no host server **18** being present.

Audio playback system **30** is an electronic device, such as a stereo receiver, AV receiver, portable stereo, amplified speaker system, or the like, that receives a digital audio stream **16** and renders it as sound over speakers **34**. The audio playback system **30** of the present example is separate from the computing device **20**, e.g. each device has its own power supply. This is not necessarily true of all embodiments. The separate computing device **20** is presumed to be within sampling range of the sound generated by audio playback system **30**, e.g. the two may be situated in the same room. The audio playback system **30** uses an audio rendering engine **32** to render sound. In the present example, the audio rendering engine **32** is presumed to have a predetermined and finite set of audio rendering capabilities, possibly due to a hardwired DSP chip comprising the engine **32**. Various other components of audio playback system **30**, such as components used to facilitate receipt the digital audio stream **16** (e.g. a network interface) and generation of sound (e.g. an amplifier), are omitted from FIG. **1** for brevity.

The speakers **34** by which sound is generated may form an integral part of the audio playback system **30** (e.g. as in a portable stereo) or may be connected to the audio playback system **30**, e.g. via speaker wire or wirelessly (as in the case of an AV receiver). In the latter case, the audio playback system **30** may have an attached or embedded Radio Frequency (RF) transmitter, and each speaker may have a complementary RF receiver. The number of speakers may vary between embodiments. For example, some audio playback system **30** may have five, six or seven speakers plus a subwoofer (referred to as a 5.1, 6.1 or 7.1 channel system).

Communications link **28** carries the digital audio stream **16** from the computer **20** to the audio playback system **30**. The communications link **28** may be virtually any form of interconnection that is capable of carrying digital information, wirelessly or otherwise, including but not limited to a wired Local Area Network (LAN) connection (e.g. Ethernet connection), Wireless LAN (e.g. WiFi™) connection, WiGig™, High-Definition Multimedia Interface (HDMI) connection, wireless HDMI, Bluetooth, WiSA, timing synchronized Ethernet protocols such as 802.1AS, a power line connection carrying data over a conductor used for electrical power transmission, optical fiber, proprietary wireless connection (e.g. AirPlay®) or the like.

Operation **200** of the computing device **20** for adjusting a data rate of a digital audio stream based on dynamically determined audio playback system capabilities is illustrated in FIG. **2**. For the purpose of this example, it is presumed that the computing device **20** is initially outputting a digital audio stream **16** to audio playback system **30** that is either uncompressed or compressed using a lossless compression technique and that the audio playback system **30** is rendering the audio stream as sound. For illustration, the digital audio stream **16** being output by the computing device **20** is presumed to have a sampling rate of 48 KHz and a bit depth (i.e. bits/sample) of 24 bits/sample, together yielding an operative data rate of 1.152 megabits per second (Mbps). The audio stream specifications of other embodiments may differ.

Initially, the computing device **20** presents a graphical user interface (GUI), such as GUI **300** of FIG. **3**, on its display (FIG. **2**, **202**). The purpose of GUI **300** is to query the user as to whether data rate adjustment is desired. The GUI **300** presented at computing device **20** may present a query such as "Check whether output audio bandwidth can be conserved with little or no loss in sound quality?" to solicit input from a user. The term "bandwidth" in the foregoing text is a colloquial reference to data rate, which may be more a familiar term for a typical user. The example GUI **300** also includes GUI controls, such as buttons **302** and **304**, for responding to the query in the affirmative or negative, respectively. The GUI **300** may be a dialog box, as illustrated in FIG. **3**, or any other type of GUI. The GUI **300** may pop up or otherwise be displayed on any number of triggering conditions, e.g. when the computing device **20** commences outputting a digital audio stream, when a user-configurable settings page or interface is invoked, when a particular application or utility is launched, or otherwise.

If the user responds in the negative to the query of GUI **300** (FIG. **2**, **204**), operation terminates. However, if the user responds in the positive, indicating a desire to proceed with the check, then another GUI may be displayed (FIG. **2**, **206**) to present the analysis to the user. An example GUI **400** for this purpose is illustrated in FIG. **4**.

Referring to FIG. **4**, GUI **400** is a dialog box including various fields and various GUI controls. A textual status field **402** is for apprising a user of the current status of the analysis. Field **402** may initially contain certain text and may be periodically updated with other text through the analysis. For example, the text in field **402** may initially be "Checking" but then may be supplemented, or replaced, with other text reflecting the ongoing analysis as it occurs, as described below.

A results field **404** provides information regarding the bandwidth conservation that is attainable, with the results being presented, e.g., as a percentage, in a text box **406**. This value could alternatively be presented as a data rate value, e.g. in Mbps or Kbps units, via a graphical indicator (e.g. a bar graph), or in some other way. The text box **406** may initially be blank, pending completion of the analysis.

A details section **408** of GUI **400** provides more specific information regarding the analytical basis for the attainable results value presented in text box **406**. The example details section **408** has three rows, labelled A), B), and C) respectively, and three columns **410**, **412**, and **414**.

Row A) indicates the characteristics of the digital audio stream **16** prior to the commencement of the bandwidth conservation analysis, i.e. before any adjustment is performed. The values in text boxes **410**A, **412**A and **414**A in columns **410**, **412** and **414**, respectively, represent the operative (pre-adjustment) sampling rate (48 KHz), operative bits/sample (24 bits/sample) and operative data rate (1.152 Mbps), respectively, of the example digital audio stream **16**. It will be appreciated that the value in text box **414**A is the product of the values in text box **410**A and text box **412**A. Row A) may be the only one of rows A)-C) that is populated prior to commencement of the analysis.

Row B) is intended to set forth the maximum usable sampling rate, maximum usable bits/sample and resultant maximum usable data rate of the audio playback system **30**. The values represent an upper threshold of digital audio stream characteristics that, if exceeded, would not result in any appreciable or significant improvement in sound quality of the sound being rendered by the audio playback system **30** and played as sound. The threshold may be due to: limitations in the audio rendering components (e.g. DSP) that are being used; limitations in the speakers through which sound is being generated by the audio playback system **30**; and/or physical characteristics of a room in which the sound is being played (e.g. anechoic quality or amount of reverberation). The three text boxes **410**B, **412**B, and **414**B are initially empty and will be populated automatically based on the outcome of sound quality sampling of the audio playback system **30** that the computing device **20** will conduct during its bandwidth conservation analysis. The value in text box **414**B will be the product of the values in text boxes **410**B and **412**B.

Row C) is will be used to set forth a recommended sampling rate, recommended number of bits/sample and resultant data rate to which the computing device **20** could reduce the digital audio stream **16** without any significant, noticeable, or possibly even any, reduction in sound quality. As will become apparent, the values in this row are based on the values in row B), but have been rounded to the closest standard values or closeby standard values. The term "standard values" includes values dictated by standards bodies or industry groups, or de facto industry standards. Depending on the embodiment, the values in row C) may be standard values that are closest to and greater than the corresponding row B) values, or closest to and less than the corresponding row B) values. The strategy that is used (i.e. greater than versus less then) in any particular embodiment may be based on which of the two competing interests of preserving sound quality and maximizing bandwidth conservation is more important in that embodiment, as will be described. The three text boxes **410**C, **412**C, and **414**C are initially empty. The value in text box **414**C will be the product of the values in text boxes **410**C and **412**C.

The GUI **400** also includes a field **416** for soliciting user input as to whether any attainable bandwidth conservation as represented in text box **406** should indeed be effected. The field includes GUI controls **418** and **420** (e.g. buttons) for indicating that adjustment should proceed or should not proceed, respectively.

In alternative embodiments, the GUI **400** may be something other than a dialog box or may comprise multiple UI pages or screens.

At the conclusion of operation **206** of FIG. **2**, the GUI **400** will be displayed. In the present example, only the text "Checking . . . " will initially appear within in field **402**. Text boxes **410**A, **412**A and **414**A will have been populated with the initial, operative data rate information of the digital audio stream **16**. All other text boxes will initially be blank.

Thereafter, the computing device **20** uses microphone **26** (FIG. **1**) to sample the sound being generated by the audio playback system **30** (FIG. **2**, **208**). In some embodiments, the computing device **20** may insert, into the digital audio stream transmitted over communications link **28**, audio data representing a particular, predetermined ("canned") sound, e.g. an impulse sound such as a chirp or sweep (whose frequency characteristics are known), to cause the audio playback system **30** to play that sound for sampling purposes. The reason is that known characteristics of that sound can be compared to whatever sound is actually generated (as sampled) in order to ascertain the quality of the generated sound.

In some embodiments, it may not be required to insert a predetermined chirp or sweep sound into the audio stream during operation **200**. Rather, it may be possible to use or manipulate the existing digital audio stream **16** for sampling purposes. This approach may entail a somewhat different, more involved analysis than that which would be undertaken for a predetermined sound.

Briefly, a short-time Fourier transform analysis (or equivalent) could be repeatedly performed on the digital audio stream and the sound generated by audio playback system **30** sampled by way of the microphone **26**. A measured time delay could be applied to enable the detected sound samples to be compared with the appropriate corresponding source samples. The source and received spectra could be monitored repeatedly or continuously until signals above a threshold (e.g., 90 dbSPL) from all frequency bins comprising the frequency spectrum, in the measureable frequency range of the microphone **26**, have been sampled. The threshold may be uniform for all frequency bins or may have differing values for different frequency bins. These sampled results would then be used for the analysis in place of the chirp. The length of time needed for arriving at a usable result by way of such an "accumulation approach" may depend on the spectral richness and/or variety of the source content of the digital audio stream **16**.

Regardless of whether a predetermined sound is used or whether the source digital audio stream **16** is used, a time delay between the output of the digital audio stream **16** by the computing device **20** and the detection of the corresponding played sound at the microphone **26** is measured. Time delay can be measured by sending out a ping and measuring the signal delay back to the microphone **26**, or by matching the envelope of the digital audio stream **16** to that of the received signal (e.g., by scanning a set of samples of the source signal and then searching the received signal, which may be stored in a buffer, for a set of samples that has a high signal correlation). Thereafter, time shift may be computed based on the known sampling rate.

The sampling in operation **208** will yield a plurality of time-domain samples, which may be stored in memory at the computing device **20**. The samples may be in LPCM format or in another format. The sampling rate may be chosen such that the Nyquist frequency is greater than the audio bandwidth for which screening is being performed, which audio bandwidth may be dictated, at least in part, by the sampling equipment (e.g. microphone **26**) that is being used. The number of bits per sample may be set to relatively high level, e.g., 18 to 24 bits per sample, in relation to common industry standard bit depths, for the sake of accuracy. The text field **402** (FIG. **4**) may be updated or supplemented with a status message such as "Sampling sound . . . " to reflect operation **208**.

Thereafter, the time-domain samples are transformed to the frequency domain, e.g. using a fast Fourier transform (FIG. **2**, **210**). For example, a high-resolution FFT (e.g., 16384 points or more) may be used for the sake of accuracy. The result will be a frequency spectrum of the sampled sound comprising a plurality of frequency components in a like plurality of frequency bins and will include amplitude and phase information regarding each of the frequency components. The amplitude information may comprise an amplitude spectrum, i.e. an amplitude value for each frequency component in the frequency spectrum. The phase information may comprise a phase spectrum, or a set of numbers showing the relative time shift of each of the different frequency components. Other representations could be used in alternative embodiments. At this stage, the text in text field **402** of GUI **400** (FIG. **4**) may be replaced with, or may be updated to read, "Computing . . . " to reflect the status of the analysis.

Using the amplitude and phase information obtained in operation **210**, the computing device **20** determines amplitude distortion and phase distortion of the sampled sound, e.g. by determining amplitude distortion and phase distortion for the frequency components of the frequency spectrum (FIG. **2**, **212**). It is presumed that the time delay has already been measured, e.g. using one of the two methods described above in conjunction with operation **208**, to allow time-matching of the frequency spectrum as sampled (representing time-delayed sound) with the corresponding portion of the digital audio stream **16**. Each bin of the frequency and phase spectra of the source audio stream (e.g., chirp or sweep) is then compared with the corresponding bin of time-matched counterpart to compute the error for each frequency bin. The amplitude and phase information, or the associated amplitude and phase distortion, may be considered to constitute or indicate a quality of the sampled sound.

To perform this analysis, first the broadband amplitude (or signal level) of the sampled frequency spectrum may be averaged and normalized against the broadband amplitude spectrum of the source signal. The spectral bins of the lower-level signal may be multiplied or scaled by a single scaling factor computed from the relative average signal (e.g., using root-mean-squared (RMS) calculations for each signal and taking the ratio) to match its overall level to that of the higher signal. Next, for each frequency bin, the measured amplitude of the normalized, received signal may be subtracted from that of the source signal. Then the absolute value of the difference may be taken, and the result divided by the amplitude of the source signal. This will yield the error for one amplitude bin of the amplitude spectrum. The same calculation may be performed on the phase bin of the phase spectrum. The spectral error on a per-bin basis may thus be obtained. This may be considered as a spectral (per-frequency bin) measurement of the amplitude and phase distortion. The resulting function of the computed error for all of the bins taken as a whole may be referred to as an error distribution.

Based on the amplitude distortion and phase distortion determined in operation **212**, the computing device **20** may then identify, within the frequency spectrum, a frequency, referred to as the upper usable frequency, above which each of the amplitude distortion and phase distortion exceed a predetermined distortion limit (FIG. **2**, **214**). The amplitude distortion limit may be separate, and may differ, from the phase distortion limit. In addition, in some embodiments, the predetermined distortion limits for amplitude and phase may vary at different frequency bins. Put another way, the predetermined amplitude distortion limit and/or the predetermined phase limit may be frequency component-specific or frequency bin-specific. That is, each component or bin may have an applicable amplitude distortion limit and/or and applicable phase limit that may differ from the amplitude distortion limit and/or phase distortion limit that is applicable to other bins or components. A subset of the frequency components of the frequency spectrum that are at or below the upper usable frequency, i.e. wherein each frequency component of the subset is less than or equal to said upper usable frequency, may be referred to as the usable frequency range. The usable frequency range is a range of frequencies within the error distribution (which will be below the Nyquist frequency of the digital sampling apparatus used to obtain the measurement) for which the per-frequency bin amplitude error and phase spectral error (as computed above) do not exceed predetermined error or distortion thresholds.

In one embodiment, the usable frequency range may be found by searching the error distribution starting from the lowest frequency, verifying that the first frequency bin falls within the error thresholds for amplitude and phase (i.e. below the applicable amplitude distortion threshold and below the applicable phase distortion threshold), and then searching upwards, bin by bin, until a first frequency bin falling outside the error thresholds (i.e. for which amplitude distortion exceeds the applicable amplitude distortion threshold and/or for which phase distortion exceeds the applicable phase distortion threshold) is found.

The usable frequency range represents the portion of the frequency spectrum, composed of frequency components of the spectrum which are at or below the upper usable frequency, that is usable by the audio playback system **30**. In other words, the frequencies within that range are the frequencies whose rendering by the audio playback system **30**, within the physical environment of the room in which the sound is being played, should result in acceptable amplitude and/or phase distortion in the generated sound.

In some embodiments, determination of the upper usable frequency may involve performing a digital room compensation analysis, e.g. generating Finite Impulse Response (FIR) correction filters for reversing room effects and linear distortion in the speakers. Techniques for performing digital room compensation analysis are known in the relevant art.

Once the upper usable frequency is known, a maximum usable sampling rate of the audio playback system is computed (FIG. **2**, **216**). The maximum usable sampling rate is a sampling rate threshold above which any increase in sampling rate of the digital audio stream provided to the audio playback system **30** would not improve the sound quality of the sound generated by the audio playback system **30**. This may be due to limited capabilities of the audio playback system **30** and/or room characteristics. In the present embodiment, the maximum usable sampling rate is computed by using the identified upper frequency limit of the usable frequency range as a Nyquist frequency. Thus the maximum usable sampling rate may be determined by doubling the upper usable frequency. For example, if the upper frequency limit were 17 KHz, the maximum usable sampling rate would be 34 KHz (samples/sec). The latter value may be populated into text box **412**A of GUI **400**.

Thereafter, the maximum usable number of bits/sample, also referred to as the maximum usable bit depth, is computed (FIG. **2**, **218**). The maximum usable bit depth is the largest bit depth usable by the audio playback system, above which added bits would not significantly or appreciably contribute to sound quality. To compute the maximum usable bit depth, a total harmonic distortion and noise (THD+N) of the usable frequency range may initially be measured. This measurement is taken with respect to the usable frequency range, rather than the entire frequency spectrum, because the frequencies above the maximum usable frequency may, by necessity (to avoid aliasing), be low-pass filtered before the digital audio stream **16** is sample-rate converted to the lower sampling rate. As such, those frequencies may be considered irrelevant.

In one example, if the original source signal has 24 bits of resolution, the maximum dynamic range will be 144 dB. The THD+N measurement might be, say, −80 dB. To compute the maximum usable number of bits per sample for this value, the THD+N measurement may be divided by a conversion factor of 6 decibels per bit, or an approximation thereof (e.g. 80 db/6 db per bit=13.3 bits/sample). This value may be populated into text box **412**B of GUI **400**.

Based on the computed maximum usable number of bits/sample and maximum usable sampling rate, the maximum usable audio data rate can be determined simply by multiplying the two together (e.g. 34K samples/second*13.3 bits/sample=452.2 Kilobits per second). This value may be populated into text box **412**C of GUI **400**.

At this stage, the maximum usable sampling rate computed in operation **216** and the maximum usable bit depth computed in operation **218** may be used to determine a reduced sampling rate and a reduced bit depth, respectively, to which the digital audio stream **16** should be adjusted or, more specifically to the present embodiment, to which the computing device **20** will recommend, in row C) of GUI **400**, that the digital audio stream **16** be adjusted contingent on user approval to proceed (FIGS. **2**, **220** and **222**).

The strategy employed for determining the recommended reduced values for the sampling rate and the bit depth may be consistent as between the two. For example, the "maximum usable" values for both parameters (i.e. for both sampling rate and the bit depth) may either be rounded up to the closest respective standard values, or they may both be rounded down to the closest respective standard values.

The rationale for adjusting the "maximum usable" values to standard values, in contrast to using the maximum usable values as such for example, is to promote compatibility with existing standards-compliant systems or technologies. Standard values may be dictated by one or more standards bodies or may be de facto industry standards. For example, in the case of sampling rates, standard values may include 8 KHz, 11.025 KHz, 16 KHz, 22.05 KHz, 32 KHz, 44.1 KHz, 47.25 KHz, 48 KHz, 50 KHz, 50.4 KHz, 88.2 KHz, 96 KHz, 176.4 KHz, 192 KHz, 352.8 KHz, 2.8224 MHz and 5.6448 MHz. In the case of bit depth, standard values may include 12, 14, 16, 18, 20 or 24 bits per sample. These examples are not intended to be exhaustive or limiting and may change as standards evolve.

The decision of whether to round the sampling rate and bit depth parameters up or down may be based on the requirements of a particular embodiment and/or user preference. A GUI control (not expressly shown) may permit entry of user input indicating whether rounding should be up or down.

For example, a decision to round the parameters up from their respective computed maximum usable values may be motivated by a desire to preserve the pre-adjustment quality of the sound being generated by the audio playback system **30** despite the reduction in the sampling rate and/or bit depth from their original values. This decision should have the effect of preserving sound quality because both of the parameters will still exceed their respective computed maximum usable values. In other words, the audio playback system **30** will still generate the best sound that it is capable of generating (at least as that sound has been sampled at the computing device **20**) despite the reduction in the sampling rate and/or bit depth from pre-adjustment values. A trade-off is that some portion of the audio information in the digital audio stream **16**, however small it may be, may effectively be "wasted" at the audio playback system **30**, in that it will not contribute to an improvement in sound quality over that which would result from the maximum usable values.

Conversely, a decision to round the maximum usable sampling rate and/or maximum usable bit depth down may be motivated by a desired to avoid the "waste" problem mentioned above, since all of the audio information in the digital audio stream **16** that is being rendered will contribute to the sound quality of the sound being generated at the audio playback system **30** in that case. It should be appreciated that this may come at the expense of a somewhat degraded sound quality. That is, the audio playback system **30** will no longer be able to generate the best sound quality that it is capable of generating (as qualified above). The reason is that the reduction in sampling rate and/or bit depth, to levels that are below the respective computed maximum usable values, will have robbed the audio playback system **30** of some of the audio information necessary to achieve that "best" sound quality.

Whichever direction of rounding is chosen (or operative by default, as may be the case for some embodiments), the reduced sampling rate is determined and automatically populated into text box **410**C, and the reduced bit depth is determined and automatically populated into text box **412**C. For example, if the direction of rounding is up, the value of 34 KHz from text box **410**B might be rounded up to a standard value of 44.1 KHz, and the value of 13.3 bits/sample from text box **412**B might be rounded up to a standard value of 14 bits/sample. The reduced data rate could thus be determined simply by multiplying the two together: 44.1 K samples/second*14 bits/sample=617.1 Kilobits per second). The latter value may be automatically populated into text box **414**C of GUI **400**. This value may be used to determine a percentage that can be automatically populated into text box **406**B. For example, presuming an original data rate of 1.152 MHz, the proposed reduced value of 617.1 Kbps would represent a bandwidth conservation of approximately 54%.

Alternatively, if the direction of rounding were down, the value of 34 KHz from text box **410**B might be rounded down to a standard value of 32 KHz, and the value of 13.3 bits/sample from text box **412**B might be rounded down to a standard value of 12 bits/sample. The reduced data rate can be determined simply by multiplying the two together: 32 K samples/second*12 bits/sample=384 Kbps), which would represent a bandwidth conservation of only approximately 33%.

Thus, when the direction of rounding of a particular embodiment is disregarded, it may be considered generally that the reduced sampling rate is a standard sampling rate selected based on closeness to the computed maximum sampling rate, and that the reduced number of bits per sample is a standard number of bits per sample selected based on closeness to the computed maximum number of bits per sample.

With the GUI **400** now being fully populated, the text "Done" may be added to, or may replace, the existing text within text field **402** to reflect the fact that the analysis is complete. At this stage, the user may elect not to proceed with the adjustment by selecting GUI control **420** (FIG. **2**, operation **224**), in which case operation **200** terminates. Alternatively, the user may elect to proceed with the adjustment by selecting GUI control **418**, in which case the recommended adjustments of row C) may be effected (FIG. **2**, operation **226**).

To effect the adjustment, the digital audio stream **16** may be format converted by the computing device **20** using any one of a number of format conversion techniques. In the case of LPCM, the sampling rate may be adjusted downwardly by applying a sample-rate conversion algorithm. The bit depth may be reduced by adding dithering at the reduced bit width and then truncating to the reduced bit width. In the case of a compression algorithm, the source (or decoded) LPCM may be re-encoded using a bit rate supported by the compression algorithm that is closest to text box **412**C of GUI **400**.

The foregoing description provides an illustration of how to perform an adjustment to a data rate of a digital audio stream **16** that is uncompressed or that is compressed utilizing a lossless compression format. If the digital audio stream **16** had been compressed using a lossy compression format, then the above-described approach may be complicated by the fact that the compression performed by computing device **20** may itself result in amplitude and/or phase distortion in the ultimately rendered sound. This may result simply from the fact that certain audio information lost in compression will not be communicated to the audio playback system **30** as part of the digital audio stream **16**. Thus it may not be possible to determine, by sampling alone, what distortion has been introduced specifically by the audio playback system **30** and/or room environment.

In some embodiments in which the digital audio stream **16** output by the computing device **20** actually originates from an upstream host server **18**, the adjustment in data rate may be applied at the host server **18** rather than the computing device **20**. For example, once the computing device **20** has presented GUI **400** and the user has indicated a desired to proceed with the adjustment, the adjusted sampling rate and bits/sample values may be communicated to the host server **18**. The host server **18** may then effect the data rate reduction upstream of the computing device **20**. This may have a benefit of freeing bandwidth in a communication link between the host server **18** and the computing device **20**. For example, network audio players such as Adobe™ Flash may support various quality levels. By default, the highest quality level at which no stuttering or dropout occurs may be selected. By using the above approach, the network audio player may be instructed to reduce its quality level based on the recommended reduced sampling rate and/or recommended reduced number of bits/sample.

Put another way, a computing device configured to output a digital audio stream to a separate audio playback system for rendering as sound over speakers may cause a data rate of the transmitted digital audio stream to be reduced, not by implementing the data rate adjustment locally, but by computing a recommended data rate reduction and communicating that information to an upstream host server **18** for implementation. This may be possible when the digital audio stream output by the computing device is based on an audio stream received from the upstream host server. A communication may be send by the computing device **20**, to the host server **18**, for causing the host server to reduce a data rate of the audio stream. The communication may include or reference the maximum sampling rate and/or the maximum usable number of bits per sample that has been computed by the computing device. The host server **18** may either reduce a sampling rate of the audio stream to a reduced sampling rate based on the communicated or referenced maximum sampling rate, or reduce a number of bits per sample of the audio stream to a reduced number of bits per sample based on the maximum usable number of bits per sample communicated or referenced by the computing device, or both. The same sorts of rounding (up or down) may be performed at the host server **18** as are described above as being performed at the computing device **20**.

As illustrated by the foregoing, adjustment of a data rate of a digital audio stream can generally be performed by sampling sound generated, from the digital audio stream, by an audio playback system and, based at least in part on a quality of the sampled sound (e.g. based on a degree of distortion detected within the sampled sound, the distortion possibly being indicative of limited audio playback system capabilities), reducing a sampling rate of the digital audio stream and/or reducing a number of bits per sample of the digital audio stream. The degree of reduction in the sampling rate or number of bits per sample be based, at least in part, upon a degree of distortion detected in the sound (e.g. the higher the distortion, the greater the reduction in data rate, generally speaking).

As will be appreciated by those skilled in the art, various modifications can be made to the above-described embodiment. For example, some embodiments may lack either or both of GUI **300** and GUI **400**. This may be the case when the computing device **20** lacks a user interface **24**. In such cases, operation **200** of FIG. **2** may commence with operation **208**, proceed to **214**, skip **216** and **218**, and end with operation **220**. In that case, operation **200** may be executed automatically ("in the background"), possibly without any awareness on the part of a user, e.g. whenever the computing device **20** commences outputting a digital audio stream **16**, whenever the computing device **20** is preparing to output other, perhaps higher priority, data over the same communications link **28** that is being used to carry the digital audio stream **16** and wishes to avoid exceeding the capacity of the link **28** or overutilizing the link **28**, at periodic intervals, or at some other logical time(s). If a predetermined (e.g. sweep) signal is being used for sampling, then activation of that signal by the user may be desired, e.g. by way of user input such as a hardware button press, because automatic insertion of such a signal may be considered obtrusive by a user. If the background accumulation approach (described above) is used, the operation could be performed while digital audio streams are being played. Once a recommended data rate has been obtained, it could be stored and applied when a new stream is started.

The above embodiment describes use of an FFT in operation **210** of FIG. **2** for performing the transformation of samples from the time domain to the frequency domain. The FFT is a mathematical optimization of the Discrete Fourier Transform (DFT), which could alternatively be used. Other alternatives for performing this transformation, such as the Modified Discrete Cosine Transform (MDCT), short-time Fourier transform, and discrete wavelet transform, could be used in some embodiments. The latter three alternatives may be more suitable than the FFT when it is desired that the analysis be localized in time as well as frequency, which may be the case when the signal is time-varying.

In the above embodiment, operation **200** is described as possibly being triggered by a triggering condition, such as upon the outputting of a digital audio stream by computing device **20**. In some embodiments, it is possible that the same, or a different, triggering condition may occur subsequent to completion of operation **200**. For example, the subsequent triggering condition may be the outputting of another different digital audio stream by computing device **20**, e.g. by a different software application than that which was responsible for outputting the first digital audio stream **16**, or some other triggering condition.

When such a subsequent triggering condition occurs, it may be possible to avoid repeating certain steps of the originally executed operation **200** when adjusting the data rate of the digital audio stream **16** to a reduced rate. For example, if it is determined, or presumed, that the setup of the audio playback system **30** and room environment is unchanged from when operation **200** was first performed, i.e. that nothing has changed that could alter the results as presented in row B), then operation **200** of FIG. **2** could skip over operations **208-218**. The GUI **400** could be presented with row A) populated with values based on the new digital audio stream **16** and row B) populated with the same values as before. Upon execution of operations **220** and **222** for the current digital audio stream **16**, row C) can be populated with new values. Subsequent operation (**224** and onward) may proceed as before. Alternatively, earlier user input might have been obtained (not expressly shown) to indicate that the data rate adjustment should be automatically performed for all digital audio streams after the first adjustment is performed. In that case, it may be possible to avoid presenting a GUI and just perform operations **220**, **222** and **226** to perform the adjustment, either without user awareness or with an indicator being displayed to show that the adjustment has been performed and to indicate the percentage or amount of the bandwidth conservation. Such automatic adjustment might also occur by default when the computing device **20** lacks a UI.

In some embodiments, a smoothing or other filtering function may optionally be applied to the error distribution before searching the distribution to identify the usable frequency range.

In some embodiments, it may be possible to reduce the data rate of the digital audio stream even lower than the value shown in text box **414**C. This may be performed by applying compression to achieve a lesser data rate, possibly with little or no reduction in perceived sound quality beyond that which would otherwise result from operation **200** of FIG. **2**. For example, the values in text boxes **414**A and **414**B (or, in some embodiments, **414**C) of FIG. **4** may be used as input to one or more lookup tables, or similar data structure(s), stored in memory at computing device **20**. The lookup table(s) may be for mapping a recommended reduced sampling rate and/or bit depth, which are in respect of an uncompressed audio stream or an audio stream compressed using lossless compression, to a compressed data stream having a lesser data rate but possibly not having significant or noticeable loss in perceived sound quality. The data structure(s) could be predetermined and may not be tailored specifically for the computing device **20** or audio playback system **30**.

For example, methods such as ABX perceptual comparison testing may be used to generate an empirical look-up table of uncompressed (e.g. LPCM) sampling rates and bit depths that are perceptually equivalent to bit rates of supported lossy compression algorithms. Such a table could be stored at the computing device **20**, e.g., using a ROM. For example, such table might rank an MP3-encoded stream with 64 kbps data rate as equivalent, or effectively equivalent, in perceptual quality to a 24 kHz, 14 bit stereo LPCM stream. Similar rankings or equivalents could be stored in the lookup table for a number of supported bit rates of a number of supported lossy compression schemes. For example, the table-lookup could be performed after step **222** to find the nearest lossy compression algorithm having an equivalent sampling rate and/or bit depth greater than or equal to the results from operations **216** and/or **218**. Once that lossy compression algorithm has been found, it may be applied to uncompressed audio at computing device **20**, and the resulting compressed audio may be transmitted to the audio playback system **30** over communications link **28**. This example presumes that the computing device **20** is itself able to apply the relevant compression. In some embodiments, the GUI **400**, or another GUI, could include a control for selectively applying such a further data rate reduction, i.e. a GUI control for selectively applying a lossy compression algorithm to an uncompressed audio stream whose data rate has already been reduced, in order to further lessen the data rate of the audio stream without significant or perceptible sound quality reduction.

It will be appreciated that the various GUI fields and/or GUI controls illustrated or described herein, e.g. in FIG. **3** or **4**, may be displayed independently from one another.

Other modifications will be apparent to those skilled in the art and, therefore, the invention is defined in the claims.

The following clauses provide a further description of example apparatuses, methods and/or machine-readable media.

1. A method of adjusting a data rate of a digital audio stream, the method comprising: sampling sound generated, from a digital audio stream, by an audio playback system; identifying a frequency above which amplitude distortion of the sampled sound exceeds a predetermined amplitude distortion limit and phase distortion of the sampled sound exceeds a predetermined phase distortion limit, the identified frequency being referred to as an upper usable frequency; computing, based on the upper usable frequency, a maximum usable sampling rate of the audio playback system; computing, based on a portion of a frequency spectrum of the sampled sound at or below the upper usable frequency, a maximum usable number of bits per sample of the audio playback system; and reducing the data rate of the digital audio stream by performing either one or both of: reducing a sampling rate of the digital audio stream to a reduced sampling rate that is determined on the basis of the computed maximum sampling rate; and reducing a number of bits per sample of the digital audio stream to a reduced number of bits per sample that is determined on the basis of on the computed maximum number of bits per sample.

2. The method of clause 1 wherein the determining of the upper usable frequency comprises: transforming the sampled sound from a time domain to a frequency domain, the transforming resulting in the frequency spectrum of the sampled sound, the frequency spectrum comprising a frequency component in each of a plurality of frequency bins, the transform-

ing yielding amplitude information and phase information regarding each of the frequency components; and using the amplitude and phase information regarding each of the frequency components, determining, for each of the frequency components of the frequency spectrum, an amplitude distortion and a phase distortion.

3. The method of clause 1 wherein the computing of the maximum usable sampling rate comprises setting the maximum usable sampling rate to twice the upper usable frequency.

4. The method of clause 1 wherein the computing of a maximum usable number of bits per sample of the audio playback system comprises: measuring a total harmonic distortion and noise (THD+N) of the portion of the spectrum at or below the upper usable frequency; and converting the THD+N measurement to the maximum usable number of bits per sample.

5. The method of clause 1 further comprising determining the reduced sampling rate of the digital audio stream by rounding the maximum usable sampling rate of the audio playback system down to the closest standard sampling rate.

6. The method of clause 1 further comprising determining the reduced sampling rate of the digital audio stream by rounding the maximum usable sampling rate up to the closest standard sampling rate.

7. The method of clause 1 further comprising determining the reduced number of bits per sample of the digital audio stream by rounding the maximum usable number of bits per sample down to the closest standard number of bits per sample.

8. The method of clause 1 further comprising determining the reduced number of bits per sample of the digital audio stream by rounding the maximum usable number of bits per sample up to the closest standard number of bits per sample.

9. The method of clause 1 wherein the identifying of the upper usable frequency comprises computing a Finite Impulse Response (FIR) filter suitable for correcting distortion resulting from either one or both of characteristics of the speakers and characteristics of a physical space in which the sound is being generated by the speakers.

10. The method of clause 1 wherein the digital audio stream comprises uncompressed audio data or audio data that has been compressed using a lossless compression format.

11. A computing device configured for outputting a digital audio stream to an audio playback system for rendering as sound over speakers, the computing device comprising a processor, the processor operable to adjust a data rate of the digital audio stream by: sampling the sound generated, from the digital audio stream, by the audio playback system; identifying a frequency above which amplitude distortion of the sampled sound exceeds a predetermined amplitude distortion limit and phase distortion of the sampled sound exceeds a predetermined phase distortion limit, the identified frequency being referred to as an upper usable frequency; computing, based on the upper usable frequency, a maximum usable sampling rate of the audio playback system; computing, based on a portion of a frequency spectrum of the sampled sound at or below the upper usable frequency, a maximum usable number of bits per sample of the audio playback system; and reducing the data rate of the digital audio stream by performing either one or both of: reducing a sampling rate of the digital audio stream to a reduced sampling rate that is determined on the basis of the computed maximum sampling rate; and reducing a number of bits per sample of the digital audio stream to a reduced number of bits per sample that is determined on the basis of on the computed maximum number of bits per sample.

12. The computing device of clause 11 wherein the processor is further operable to determine the reduced sampling rate of the digital audio stream by rounding the maximum usable sampling rate of the audio playback system down to the closest standard sampling rate.

13. The computing device of clause 11 wherein the processor is further operable to determine the reduced sampling rate of the digital audio stream by rounding the maximum usable sampling rate up to the closest standard sampling rate.

14. The computing device of clause 11 wherein the processor is further operable to determine the reduced number of bits per sample of the digital audio stream by rounding the maximum usable number of bits per sample down to the closest standard number of bits per sample.

15. The computing device of clause 11 wherein the processor is further operable to determine the reduced number of bits per sample of the digital audio stream by rounding the maximum usable number of bits per sample up to the closest standard number of bits per sample.

16. A tangible machine-readable medium storing instructions that, upon execution by a processor of a computing device, the computing device configured to output a digital audio stream to an audio playback system for rendering as sound over speakers, cause the processor to: sample the sound generated, from the digital audio stream, by the audio playback system; identify a frequency above which amplitude distortion of the sampled sound exceeds a predetermined amplitude distortion limit and phase distortion of the sampled sound exceeds a predetermined phase distortion limit, the identified frequency being referred to as an upper usable frequency; compute, based on the upper usable frequency, a maximum usable sampling rate of the audio playback system; compute, based on a portion of a frequency spectrum of the sampled sound at or below the upper usable frequency, a maximum usable number of bits per sample of the audio playback system; and reduce the data rate of the digital audio stream by performing either one or both of: reducing a sampling rate of the digital audio stream to a reduced sampling rate that is determined on the basis of the computed maximum sampling rate; and reducing a number of bits per sample of the digital audio stream to a reduced number of bits per sample that is determined on the basis of on the computed maximum number of bits per sample.

17. The machine-readable medium of clause 15 wherein the instructions further cause the processor to determine the reduced sampling rate of the digital audio stream by rounding the maximum usable sampling rate of the audio playback system either down to the closest standard sampling rate or up to the closest standard sampling rate.

18. The machine-readable medium of clause 15 wherein the processor wherein the instructions further cause the processor to determine the reduced number of bits per sample of the digital audio stream by rounding the maximum usable number of bits per sample either down to the closest standard number of bits per sample or up to the closest standard number of bits per sample.

19. The tangible machine-readable medium of clause 16 wherein the instructions further cause the processor to: display, at the computing device, a graphical user interface (GUI) comprising at least one of: an indication of the computed maximum usable sampling rate of the audio playback system; and an indication of the computed maximum usable number of bits per sample of the audio playback system.

20. The tangible machine-readable medium of clause 16 wherein the instructions further cause the processor to: display, at the computing device, a graphical user interface (GUI) comprising at least one of: an indication of the reduced

sampling rate that has been determined on the basis of the computed maximum usable sampling rate of the audio playback system; and an indication of the reduced number of bits per sample that has been determined on the basis of the computed maximum usable number of bits per sample of the audio playback system.

What is claimed is:

1. A method of adjusting a data rate of a digital audio stream, the method comprising:

sampling sound generated, from a digital audio stream, by an audio playback system;

ascertaining, from the sampled sound, a sound quality of the generated sound; and

based at least in part on the ascertained sound quality of the generated sound, reducing the data rate of the digital audio stream by performing either one or both of:

reducing a sampling rate of the digital audio stream to a reduced sampling rate; and

reducing a number of bits per sample of the digital audio stream to a reduced number of bits per sample.

2. The method of claim **1**, further comprising:

identifying a frequency above which amplitude distortion of the sampled sound exceeds a predetermined amplitude distortion limit and phase distortion of the sampled sound exceeds a predetermined phase distortion limit, the identified frequency being referred to as an upper usable frequency; and

computing, based on the upper usable frequency, a maximum usable sampling rate of the audio playback system,

wherein the reduced sampling rate is determined on the basis of the computed maximum usable sampling rate.

3. The method of claim **2** wherein the determining of the upper usable frequency comprises:

transforming the sampled sound from a time domain to a frequency domain, the transforming resulting in the frequency spectrum of the sampled sound, the frequency spectrum comprising a frequency component in each of a plurality of frequency bins, the transforming yielding amplitude information and phase information regarding each of the frequency components; and

using the amplitude and phase information regarding each of the frequency components, determining, for each of the frequency components of the frequency spectrum, an amplitude distortion and a phase distortion.

4. The method of claim **2** wherein the computing of the maximum usable sampling rate comprises setting the maximum usable sampling rate to twice the upper usable frequency.

5. The method of claim **2** further comprising determining the reduced sampling rate of the digital audio stream by rounding the maximum usable sampling rate of the audio playback system down to a closest standard sampling rate.

6. The method of claim **2** further comprising determining the reduced sampling rate of the digital audio stream by rounding the maximum usable sampling rate up to a closest standard sampling rate.

7. The method of claim **1**, further comprising:

identifying a frequency above which amplitude distortion of the sampled sound exceeds a predetermined amplitude distortion limit and phase distortion of the sampled sound exceeds a predetermined phase distortion limit, the identified frequency being referred to as an upper usable frequency; and

computing, based on a portion of a frequency spectrum of the sampled sound at or below the upper usable frequency, a maximum usable number of bits per sample of the audio playback system,

wherein the reduced number of bits per sample is determined on the basis of the computed maximum usable number of bits per sample.

8. The method of claim **7** wherein the determining of the upper usable frequency comprises:

transforming the sampled sound from a time domain to a frequency domain, the transforming resulting in the frequency spectrum of the sampled sound, the frequency spectrum comprising a frequency component in each of a plurality of frequency bins, the transforming yielding amplitude information and phase information regarding each of the frequency components; and

using the amplitude and phase information regarding each of the frequency components, determining, for each of the frequency components of the frequency spectrum, an amplitude distortion and a phase distortion.

9. The method of claim **7** wherein the computing of a maximum usable number of bits per sample of the audio playback system comprises:

measuring a total harmonic distortion and noise (THD+N) of the portion of the spectrum at or below the upper usable frequency; and

converting the THD+N measurement to the maximum usable number of bits per sample.

10. The method of claim **7** further comprising determining the reduced number of bits per sample of the digital audio stream either by rounding the maximum usable number of bits per sample down to a closest standard number of bits per sample or by rounding the maximum usable number of bits per sample up to a closest standard number of bits per sample.

11. The method of claim **1** wherein the digital audio stream comprises uncompressed audio data or audio data that has been compressed using a lossless compression format.

12. A computing device configured for outputting a digital audio stream to an audio playback system for rendering as sound over speakers, the computing device comprising a processor, the processor operable to adjust a data rate of the digital audio stream by:

sampling the sound generated, from the digital audio stream, by the audio playback system;

ascertaining, from the sampling, a sound quality of the generated sound; and

based at least in part on the ascertained sound quality of the generated sound, reducing the data rate of the digital audio stream by performing either one or both of:

reducing a sampling rate of the digital audio stream to a reduced sampling rate; and

reducing a number of bits per sample of the digital audio stream to a reduced number of bits per sample.

13. The computing device of claim **12** wherein the processor is further operable to:

identify a frequency above which amplitude distortion of the generated sound exceeds a predetermined amplitude distortion limit and phase distortion of the generated sound exceeds a predetermined phase distortion limit, the identified frequency being referred to as an upper usable frequency; and

compute, based on the upper usable frequency, a maximum usable sampling rate of the audio playback system,

wherein the reduced sampling rate is determined on the basis of the computed maximum usable sampling rate.

14. The computing device of claim **13** wherein the processor is further operable to determine the reduced sampling rate of the digital audio stream either by rounding the maximum usable sampling rate of the audio playback system down to a closest standard sampling rate or by rounding the maximum usable sampling rate up to a closest standard sampling rate.

15. The computing device of claim **12** wherein the processor is further operable to:

identify a frequency above which amplitude distortion of the generated sound exceeds a predetermined amplitude distortion limit and phase distortion of the generated sound exceeds a predetermined phase distortion limit, the identified frequency being referred to as an upper usable frequency; and

compute, based on a portion of a frequency spectrum of the generated sound at or below the upper usable frequency, a maximum usable number of bits per sample of the audio playback system,

wherein the reduced number of bits per sample is determined on the basis of the computed maximum usable number of bits per sample.

16. The computing device of claim **15** wherein the processor is further operable to determine the reduced number of bits per sample of the digital audio stream either by rounding the maximum usable number of bits per sample down to a closest standard number of bits per sample or by rounding the maximum usable number of bits per sample up to a closest standard number of bits per sample.

17. A non-transitory tangible machine-readable medium storing instructions that, upon execution by a processor of a computing device, the computing device configured to output a digital audio stream to an audio playback system for rendering as sound over speakers, cause the processor to:

sample the sound generated, from the digital audio stream, by the audio playback system;

ascertaining, from the sampling, a sound quality of the generated sound; and

based at least in part on the ascertained sound quality of the generated sound, reduce the data rate of the digital audio stream by performing either one or both of:

reducing a sampling rate of the digital audio stream to a reduced sampling rate; and

reducing a number of bits per sample of the digital audio stream to a reduced number of bits per sample.

18. The machine-readable medium of claim **17** wherein the instructions further cause the processor to:

identify a frequency above which amplitude distortion of the generated sound exceeds a predetermined amplitude distortion limit and phase distortion of the generated sound exceeds a predetermined phase distortion limit, the identified frequency being referred to as an upper usable frequency;

compute, based on the upper usable frequency, a maximum usable sampling rate of the audio playback system; and

compute, based on a portion of a frequency spectrum of the sampled sound at or below the upper usable frequency, a maximum usable number of bits per sample of the audio playback system,

wherein the reduced sampling rate is determined on the basis of the computed maximum usable sampling rate, and

wherein the reduced number of bits per sample is determined on the basis of the computed maximum usable number of bits per sample.

19. The machine-readable medium of claim **18** wherein the instructions further cause the processor to:

display, at the computing device, a graphical user interface (GUI) comprising at least one of:

an indication of the computed maximum usable sampling rate of the audio playback system; or

an indication of the computed maximum usable number of bits per sample of the audio playback system.

20. The machine-readable medium of claim **18** wherein the instructions further cause the processor to:

display, at the computing device, a graphical user interface (GUI) comprising at least one of:

an indication of the reduced sampling rate that has been determined on the basis of the computed maximum usable sampling rate of the audio playback system; and

an indication of the reduced number of bits per sample that has been determined on the basis of the computed maximum usable number of bits per sample of the audio playback system.

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